Listing of Claims:

1. (Previously Presented) A method for evaluating a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, said method comprising the steps of:

obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received packets of the audio stream;

duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time.

2. (Previously Presented) The method according to claim 1, wherein the determined time difference between said first and second speech signals is measured by intercorrelation of envelope signals of said first and second speech signals.

3. (Previously Presented) The method according to claim 1, wherein the step of determining the time difference is preceded by a step of detecting vocal activity in the first and second speech signals, the determining and calculating steps being executed if vocal activity detected in the first and second signals is above a predetermined threshold.

4. (Previously Presented) The method according to claim 1, wherein said decoding within a predetermined decoding time implements one of a decoding algorithm identical to that implemented in said telephony module and a constant and known decoding time difference relative to the algorithm implemented in the telephony module.

5. (Previously Presented) The method according to claim 1, wherein the processing delay is obtained by summing the determined time difference between the first and second speech signals and the predetermined decoding time of the first speech signal.

6. (Previously Presented) The method according to claim 1, wherein said packet switching network is an IP network and the data packets received in the terminal are IP packets.

7. (Previously Presented) A method according to claim 1, further comprising the step of:

evaluating the calculated processing delay of the speech signal in the terminal to evaluate end-to-end transmission delay of the speech signal contained in the data packets received in the receiver terminal during the voice call to the terminal sending said speech signal over the packet-switched network.

8. (Previously Presented) The method according to claim 7, further comprising:

evaluating the processing delay of the speech signal sent over the packet-switched network;

measuring the transmission delay of the speech signal in the packet-switched network; and

evaluating the end-to-end transmission delay from said processing delay of the speech signal sent over the packet-switched network, said transmission delay of the speech signal in the packet-switched network and said processing delay of the speech signal received in the receiver terminal.

9. (Previously Presented) The method according to claim 8, wherein the processing delay of the speech signal sent over the packet-switched network is evaluated by consulting a table stored in the receiver terminal containing a predefined maximum value and a predefined minimum value of said processing delay of the speech signal sent over the packet-switched network for each type of speech signal send coder, predefined maximum values accounting for payload of received IP packets.

10. (Previously Presented) The method according to claim 8, wherein the transmission delay of the speech signal in the packet-switched network is evaluated using a Ping technique.

11. (Previously Presented) The method according to claim 8, wherein the transmission delay of the speech signal in the packet-switched network is evaluated from sender report information extracted from the received data packets.

12. (Previously Presented) The method according to claim 7, wherein the end-to-end transmission delay is evaluated by summing send processing delay of the speech signal sent over the packet-switched network, said transmission delay of the speech signal in the packet-switched network and said processing delay of the speech signal received in the receiver terminal.

13. (Previously Presented) The method according to claim 7, further comprising the steps of:

creating information representing obtained end-to-end delay values; and

sending said created end-to-end delay information over the packet-switched network to a collection server configured to manage end-to-end delay information sent by a plurality of communication terminals connected to the network.

14. (Previously Presented) A device for evaluating a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, said device comprising:

a network filter module configured to obtain, at the receiver terminal, a stream of audio packets from the received data packets;

a control decoder module having a predetermined decoding time for decoding the stream of audio packets obtained and for reconstituting a first speech signal from the received packets of the audio stream;

an audio filter module configured to duplicate, at the receiver terminal, at

least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal;

means for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

means for calculating, at the receiver terminal, the processing delay of the speech signal contained in data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and the predetermined decoding time.

15. (Previously Presented) The device according to claim 14, wherein the time difference between the first speech signal and the second speech signal is measured by intercorrelation of the envelope signals of first and second speech signals.

16. (Previously Presented) The device according to claim 14, further comprising:

means for evaluating the calculated processing delay of the speech signal contained in data packets received in the terminal to evaluate end-to-end transmission delay of the speech signal during the voice call to the terminal sending said data packets over the packet-switched network, said evaluating means being configured for installation the receiver terminal having the telephony module.

17. (Previously Presented) Telephone terminal equipment on a packet-switched network, said telephone terminal equipment including a device for evaluating the processing delay of a speech signal as claimed in claim 14.

18. (Previously Presented) Telephone terminal equipment on a packet-switched network, said telephone terminal equipment including a device for evaluating the end-to-end transmission delay of a speech signal as claimed in claim 16.

19. (Currently Amended) A computer-readable ~~information~~ storage medium encoded with a computer program executed by a computer that causes evaluation of a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, the computer program comprising:

program code for obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received packets of the audio stream;

program code for duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

program code for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

program code for calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time.

20. (Currently Amended) A computer-readable ~~information~~ storage medium encoded with a computer program executed by a computer that causes evaluation of a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, the computer program comprising:

program code for obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received packets of the audio stream;

program code for duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

program code for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

program code for calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time; and

program code for evaluating the calculated processing delay of the speech signal in the terminal to evaluate end-to-end transmission delay of the speech signal contained in the data packets received in the receiver terminal during the

voice call to the receiver terminal sending said speech signal over the packet-switched network.

21. (Previously Presented) The device according to claim 14, further comprising:

means for detecting vocal activity in the first and second speech signals, the time difference between the first and second speech signals being determined if detected vocal activity is above a predetermined threshold.

22. (Previously Presented) The telephone terminal equipment on a packet-switched network as clamed in claim 17, wherein said telephone terminal equipment comprises an IP telephone or a personal computer having telephony software.

23. (Previously Presented) The telephone terminal equipment on a packet-switched network as clamed in claim 18, wherein said telephone terminal equipment comprises an IP telephone or a personal computer having telephony software.